RESOUNDIFY

VoxNova128F

Open Structure DSP 12X8, Dante 64X64, 8 flex channels, 12AEC

PRODUCT OVERVIEW

The **Resoundify VoxNova128F** is a next-generation audio DSP processor designed for advanced AV, conferencing, and networked audio environments. With 12 analog inputs, 8 analog outputs, and 64x64 Dante™ digital audio networking, it delivers scalable, crystal-clear audio performance for demanding applications. Equipped with 12 channels of full-duplex Acoustic Echo Cancellation (AEC) and 8 Flex Channels (user-assignable as AEC, line in/out, or virtual processing), the VoxNova128F provides unmatched adaptability in voice and video conferencing systems.

Built on a professional SHARC DSP platform, this unit offers open architecture signal flow customization, powerful matrix mixing, and real-time adaptive processing. The VoxNova128F is the ideal backbone for environments that require dynamic audio routing, high channel density, and superior clarity — including boardrooms, command centers, hybrid learning spaces, and corporate AV networks.

KEY FEATURES

- **Professional SHARC DSP Core:** Built on the ADI SHARC platform with a semi-open architecture, the VoxNova128F delivers exceptional signal processing power and flexibility for custom audio flow designs and advanced configurations.
- **High-Quality Audio Processing:** 24-bit/48kHz audio resolution ensures crystal-clear sound quality across all channels.
- **Intelligent Feedback Suppression:** Independent adaptive feedback suppression on each channel automatically eliminates unwanted noise.
- **Full-Duplex AEC & ANC:** Integrated Adaptive Echo Cancellation and Active Noise Cancellation for clear communication in conferencing environments.
- **Auto Mixer & Gain Control:** Built-in Gain Sharing Auto Mixer, Automatic Gain Control (AGC), and Audio Ducking (Ducker) for seamless level balancing.
- **Ambient Noise Compensation:** Real-time Ambient Noise Compensator (ANC) adjusts audio levels based on environmental sound.
- **Comprehensive Audio Matrix:** Flexible mixing matrix with input level control, channel duplication, linking, and grouping.
- **Expandable Control Options:** 8 configurable GPIOs (input/output/ADC), RS-232 & UDP support with assignable ports for central control systems.
- **Multi-Platform Compatibility:** Supports both iOS and Windows OS with dual USB audio interface for recording and conferencing.

APPLICATIONS

- Boardrooms
- Classrooms
- Auditorium

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TECHNICAL SPECIFICATIONS

System Specifications

Processor	ADI SHARC 21569@1GHz SIMD*2
Raw Processing Capacity	500 MIPS, 6 GFLOPS, 2 GMACS
Sampling Rate	48 kHz ± 100 ppm
Frequency Response (A/D/A)	20 Hz - 20 kHz ± 0.5 dB
Dynamic Range (A/D/A)	117 dB (A-weighted)
THD + Noise	<0.003%@4dBu
Channel Separation (A/D/A)	108 dB @ 1 kHz, +24 dBu
Latency (A/D/A)	<1 ms (input routed directly to output)
Delay Memory	174 mono seconds
Analog Control Inputs	0-3.3 VDC
Recommended External Control Potentiometer	10k Ohm, linear taper
Logic Outputs	Low (0 V) when active Pulled high (5 V) when inactive
Logic Output Maximum External Power Supply / Current Sinking	24 VDC / 50 mA
Logic Output Maximum Output Current	10 mA
RS-232 Accessory Serial I/O	57.6 kbps (default), 8 data bits, 1 stop bit, no parity, Straight-through wiring; pins 2, 3, 5 used
Maximum Stored Presets	1,000 storable presets

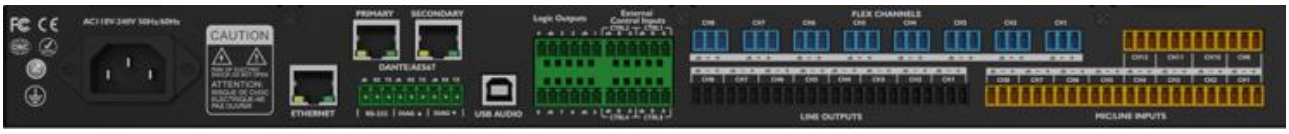
Analog Inputs and Outputs

Number of Analog Inputs	12 switchable balanced mic or line level
Analog Input and output Connectors	3.81 mm terminal blocks
Nominal Analog Input and output Level	+4 dBu with 20 dB headroom
Analog Input and output Maximum Level	+24 dBu (or +22.8 dBu into a 2k Ohm minimum load)
Analog Mic Pre-amp Gain	0 to 51 dB (in 3 dB steps) with ±24 dB digital trim
Analog Mic Pre-amp EIN	< -125 dB (with 150 Ohm source, 22.4 kHz BW)
Analog Input Impedance	5.4k Ohms balanced, 1k Ohms unbalanced
Analog Phantom Power (per input)	+48 VDC per input, max 10 mA
Analog Input Dynamic Range	>117 dB, A-weighted
Analog Input THD + Noise	<-100 dB (22.4 kHz BW, unweighted), 1 kHz @ +15 dBu, 0 dB gain
Analog Input Latency	0.30 ms
Number of Analog Outputs	8 balanced line level
Analog Output Impedance	600 Ohms balanced, 300 Ohms unbalanced
Analog Output Dynamic Range	117 dB, A-weighted
Analog Output THD + Noise	< -97 dB (22.4 kHz BW, unweighted); 1 kHz, 0 dB gain, +8 dBu output
Analog Output Latency	0.30 ms

AEC8

AEC Number of Channels	12 Channels
AEC Tail Length	512 ms - suitable for medium rooms
AEC Convergence Rate	Typically > 90 dB/sec
AEC Latency	16 mS
AEC Processors	ADI SHARC 21569@1GHz

Rear View



Control Software

VoxControl+ is our dedicated configuration software, available for free download from our official website. Designed with a user-friendly interface, it allows fine-tunusers to easily tailor the matrix settings to match the specific needs of any installation. With this software, you can e a wide range of parameters, including:

- Input gain
- Expander
- Compressor & Limiter
- Auto Gain Control (AGC)
- Equalizer
- Figure Balancer
- Active Noise Control (ANC)
- Feedback (AFC)
- Noise gate
- Ducker
- SPL
- Share AM (Automixer)
- Echo Canceller (AEC)
- Camera Tracking
- Noise Suppresion (ANS)
- Matrix
- Low & High Pass filters
- Delayer
- Output